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Performance Analysis of Voice Quality aspects in a Wireless LAN Network

Mohammed Wasif
HCL Technologies Ltd, Chennai, Tamil Nadu, India
R.K.Singh
Bharat Sanchar Nigam Ltd, Ludhiana, Punjab, India
Email: md.wasif@hcl.com,rajiv_bhu@gmail.com

(Abstract) Voice over Internet Protocol (VoIP) is a rapidly evolving technology that could possibly revolutionize the Telecommunication industry. VoIP over a wireless local area network (WLAN) is poised to become an important Internet application. When implemented on wireless data networks, VoIP could prove to be instrumental in the convergence of existing fixed and cellular telephony networks with the fast growing wired and wireless data networks. In this paper, we discuss the methodology and results of our experiments to measure throughput, delay and loss over packet networks. Measurements were performed over 802.11b networks in both ad-hoc and infra-structure modes and focused on the three major VoIP metrics: loss, delay and jitter. This system is distinguished from the existing tools that define only the nature of voice traffic, process those packets in the same way as general data, and analyze the quality of packet transmission such as loss, delay, jitter. Here different models for loss and estimation of parameters using trace files have been done. Experimental data was recorded in a repository for further study.

Keywords: Ad-hoc; Infrastructure; QoS; Performance; Wireless LAN; VoIP

1. INTRODUCTION

Voice over Internet Protocol (VoIP) has been envisioned as the next likely revolution in the telecommunications and computer network industry . VoIP involves digitization of voice streams and transmitting the digital voice as packets over conventional IP data networks like the Internet. Wireless Local Area Networks (WLANs) have made impact inside and outside the enterprise environment. Increasingly, corporations are turning towards WLANs their offices and campuses. Similarly, the general consumers are increasingly installing WLAN equipment homes to network their home computers and appliances. With the rapid proliferation of wireless LANs (WLANs) operating at high data rates up to 54 Mbps , VoIP deployed over such networks could provide an economical alternative to conventional cellular telephony. VoIP will eventually result in the convergence of existing data and telephony networks. The existing Public Switched Telephone Network (PSTN), the cellular network, the wired Internet and the WLANs could possibly converge forming a single network offering multiple services.

VoIP technology is still in the early stages of commercial deployment over wireless LANs.. It is essential to determine the number of simultaneous users a wireless network can support simultaneously without significantly degrading the QoS .The major factors affecting QoS in a wireless network are throughput, packet loss, packet delays and jitter. The wireless channel changes with time and hence it is possible that a link between two nodes could break in midst of a voice session. This makes it difficult to achieve an acceptable QoS in a wireless environment.The codecs used for encoding and decoding the voice also impact the QoS. 802.11b provides wireless connectivity with speeds of up to 11Mbps.The main goal of this project is to analyze the 4 main QoS (Quality of Service) measurements in a Voice over 802.11b system. They are end-to-end delay, delay-jitter, throughput and packet loss. The parameters in a Voice over 802.11b network which affect these QoS values include the connection speed of the network, the type of voice encoding chosen, and the number of stations nodes in the network.

2. BACKGROUND

A. VoIP Attributes

Good quality of service means providing satisfactory experience to the end users. Voice quality is related to three major factors: delay, loss, and jitter. While data traffic is mostly affected by packet loss and is more resilient to delay, voice calls are loss tolerant but more sensitive to delay and jitter (delay variance). In voice communication delay usually refers to the end-to-end delay, that is the time a packet takes to travel from the sender process to the receiver process. End-to-end delay has a significant effect on the perceived quality of IP telephony, and it can be the case that it is not the same in both directions [1]. Packet delay, which is a measure of the time taken by a packet to travel from an originating node to a destination node.The second parameter is packet jitter, which is a measure of delay variability. For this purpose we measured the time between

packet arrivals. We checked the time variability between arrivals, because for jitter the average is not important but the variance is. The last parameter is lost packet rate. This is a measure of how many packets are dropped in a particular conversation.

B. IEEE 802.11b

802.11b Wireless Networks 802.11b is a standard published by the IEEE specifying the MAC (Medium Access Control) and PHY (Physical Layer) procedures of a wireless local area network. The basic principle behind 802.11b is using the 2.4GHz frequency range as a shared medium between wireless stations. Each wireless station employs a CSMA/CA scheme that allows the stations to communicate with each other on the shared medium. [2]

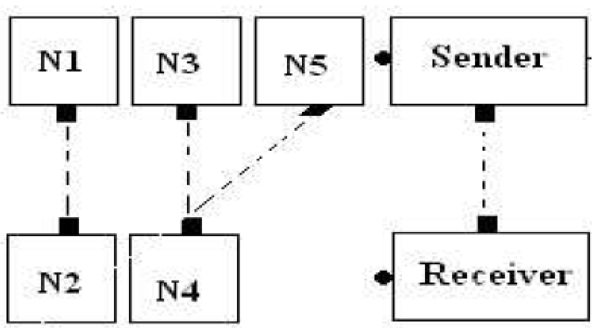

Figure 1. Basic Infrastructure mode Setup

802.11b networks offer two modes of operation: Infrastructure and Ad-hoc. In an infrastructure network (Figure 1), an AP (Access Point) [3] serves as an administrative node controlling admission of stations onto the network. In Ad-hoc mode (Figure 2), all stations are free to communicate with each other as peers.

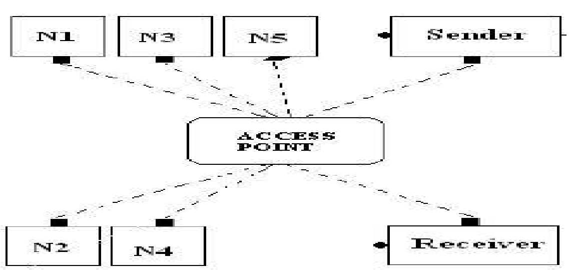

Figure 2. Basic Ad-hoc mode Setup

3. METHODOLOGY

A. VoIP Traffic

First of all, we needed to decide on the VoIP traffic mechanism because there are many different types. Our choice was to use a 64Kbps half-duplex UDP flow, which means that only the node which established the call sends traffic at first. Then, after a random time, this node stops and the node which received the call starts to send traffic after a configurable blank time space. The time each node spends sending traffic was randomised. Usually, the node which sent traffic changed several times during the simulation. To generate a 64 Kbps flow, 512 bytes packets were sent every .064 milliseconds.

B. Events Simulated

Four events were simulated. One under an Infrastructure mode architecture and the other three under an Ad-hoc mode network. They are **:**

- Infrastructure real mode. 20 nodes plus AP (Figure 3).
- Ad-hoc real mode. 21 nodes (Figure 4).
- Ad-hoc real mode. 31 nodes.
- Ad-hoc real mode. 41 nodes.

The key point in the infrastructure mode is the AP. When more conversations are added, the AP has to schedule more packet transmissions hence its packet queue will lengthen. These packets which are waiting to be forwarded will be delayed. The AP queue has a limited capacity and when the queue is full, if a new one is received it will be dropped. The AP queue has a limited capacity and when the queue is full, if a new one is received it will be dropped. In the ad-hoc case, a routing protocol has to establish routes among nodes in order to establish end to end user conversations. In this case, there will normally be intermediate nodes in a call. Depending on which nodes establish a VoIP call there will be more or less hops needed for a packet to arrive at the destination node. It is possible that some node in a conversation path will become saturated, because it could be a node involved in many conversations. In this case, all the conversations which use that node will be affected, their packets will be delayed and, if the node packet queue is full, some packets will be dropped. This is the normal behavior in an ad-hoc network.

In the infrastructure mode to obtain results it is enough to analyze one call to understand what is happening in that scenario, even if more calls are running at the same time. Since there is a centralized control element, the AP, and there is no QoS mechanism running, all packets receive the same treatment. Therefore, when the AP queue is saturated, packets from all conversations are dropped on a random basis, and packets from all conversations are delayed in the same way. Knowing this, if one conversation starts to present bad results in delay, or dropped packets appear for that conversation, the rest of conversations will also present the same behaviour and bad quality and it is enough to check only one conversation. in a particular simulation.

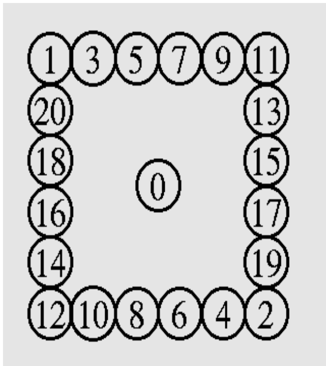

Figure 3. Infrastructure Real mode Setup

In the ad-hoc real mode we had to obtain data for every VoIP call running, because in this case

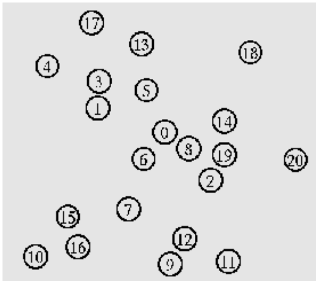

Figure 4. Ad-hoc Real mode Setup (21 Nodes)

every call has a different path and different number of hops.

4. RESULTS AND DISCUSSION

The following are the results of simulating VoIP traffic which includes Infrastructure mode setup, Ad-hoc mode setup with 3 different scenarios. All the packet information is directly derived from the trace files and plotted using gnuplot. The simulation environment used here for all the results obtained is by Network Simulator. Here in the results we first plot throughput of both send and receive packets versus simulation time for all the three cases in Ad-hoc mode and for the Infrastructure mode as shown below. We observe that as the number of nodes increases the delay (given by send throughput - receive throughput) also increases. This is because as the number of nodes increases the voice calls and packet data has to pass through various nodes along the route before reaching destination node. So all the intermediate nodes have to transmit its own packets which are scheduled for transmission and also the packets forwarded by other nodes, as a result the density of packet transfer will be more which in turn leads to more delay with increase in number of nodes.

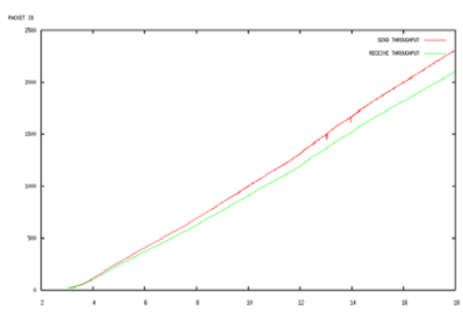

Figure 5. Ad-hoc mode (21 Nodes) Delay graph

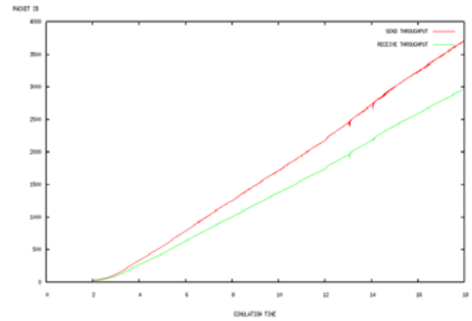

Figure 6. Ad-hoc mode (31 Nodes) Delay graph

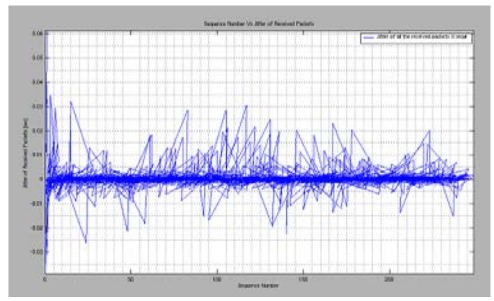

Figure 7. Ad-hoc mode (21 Nodes) Jitter graph

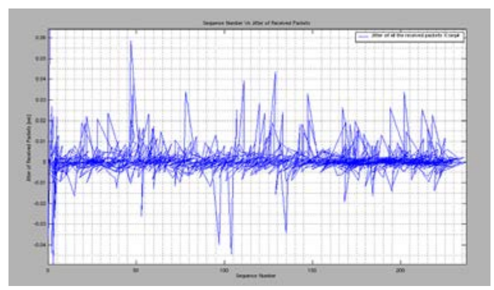

Figure 8. Ad-hoc mode (31 Nodes) Jitter graph

The next parameter is packet receive time at destination node and its relationship with end-to-end delay. In voice communication delay usually refers to the end-to-end delay, which is the time a packet takes to travel from the sender process to the receiver process. Here we plotted packet receive time at destination node versus end-to-end delay for Ad-hoc mode setup. We can observe that as the number of nodes increases for the scenarios created, the end-to-end delay also increases because as the number of nodes increases there is a chance of congestion being more in the network.

5. CONCLUSION AND FUTURE WORK

In this paper we have evaluated the performance of VoIP calls over 802.11b WLANs, in Infrastructure and ad-hoc mode architectures. We measured different parameters, involved in every real-time communication, such as packet delay, jitter and lost packet rate, and on the basis of them we have determined the VoIP performance. A statistical analysis can also be done to enable us to look for

comparison patterns. In this way just by making a comparison with a given distribution we can know, approximately, what is happening in that conversation. We conclude that using an ad-hoc link, voice quality decreases with the distance and losses are the major impediment to voice quality.

It is nearly impossible to cover all the possible measurement and analysis cases. Therefore we list in our priority order some tasks that would be natural follow from the work presented. Since quality of 802.11b networks is our main focus it would be sensible to combine the loss, delay and jitter tuple into a single value. Some research on how to combine the quality tuple into a single value exists and to report this would be useful. Most people, however, understand loss, delay and jitter better than a single value. This could also be integrated using any real time tool.

6. References

[1]. Wei Wang, Soung Chang Liew, Victor O. K. Li, "Solutions to Performance Problems in VoIP over a 802.11 Wireless LAN", IEEE Transactions on Vehicular Technology, Vol. 54, No. 1, pp. 366-384, Jan. 2005.

[2]. Kamesh Medepalli, Praveen Gopalakrishnan, David Famolari and Toshikazu Kodama, "Voice Capacity of IEEE 802.11b, 802.11a and 802.11g Wireless LANs", IEEE Communications Society, pp. 1549-1553, Globecom 2004.

[3]. David P. Hole, Fouad A. Tobagi, "Capacity of an IEEE 802.11b Wireless LAN supporting VoIP", Proceedings. IEEE International Conference on Communications (ICC), 2004.

[4]. Sachin Garg, Martin Kappes, "An Experimental Study of Throughput for UDP and VoIP Traffic in IEEE 802.11b Networks", IEEE Journal on Selected Areas in Communications, pp. 1748-1753, 2004.

[5]. Moncef Elaoud, David Famolari, Ahbrajit Ghosh, "Experimental VoIP Capacity Measurements for 802.11b WLANs" IEEE Journal on Selected Areas in Communications, pp. 272-277, 2004.

[6]. Andreas Kopsel, Adam Wolisz, "*Voice transmission in an IEEE 802.11 WLAN based access network*", *ACM Journal*, pp. 23-32, 2001.

AUTHOR INTRODUCTION

Author1:

Mohammed Wasif is a Technical Leader in HCL Technologies Ltd, India and is currently leading a project for Juniper Networks Ltd. Before joining HCL Technologies Ltd, he worked as an IT Analyst in Tata Consultancy Services Ltd, India. He completed his Master of Technology (M.Tech) in Communication Engineering from VIT University, Vellore, India in 2004 and did his Bachelor of Engineering (B.E) in Electronics and Communication Engineering discipline at M.J.P Rohilkhand University, Bareilly, India in 2002. He has published Research papers in peer-reviewed international journal and in IEEE international conferences and his research interests include VLSI for Wireless Communication Systems, MIMO Channel, Network-On-Chip (NOC), SDR, VOIP, IMS, Mobile Ad-Hoc and High-Performance Computer Networks.

Author2 :

Dr. R. K. Singh was born in Jaunpur, India in 1978. He received his Ph.D. degree in Electronics Engineering from Indian Institute of Technology (BHU), Varanasi in 2011. He is currently with Bharat Sanchar Nigam Ltd and is responsible for installation, commissioning, operation and maintenance of WiMAX service network in Ludhiana SSA. He was recipient of Senior Research Fellowship at Centre of Advanced Study, Deptt. of Electronics Engg., IT-BHU, Varanasi. He has authored and co-authored more than 25 research papers in peer-reviewed national/international journals and conference proceedings including IEEE. His current research interests include wired and wireless technologies; electromagnetic radiation; analysis, design and simulation of high-power high-frequency microwave devices and systems for communication purposes.